Essence ®
Technology

EIP 7012/EIP 7012P

Essence VoIP Phone User's Manual

Version 1.0D

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1 Introduction

This administration guide is for EIP 7012. The administration guide explains the keypad instruction, web configuration and command line configuration for VoIP Phone. Before using VoIP Phone, some setup processes are required to make the VoIP Phone work properly. Please refer to the Setup Menu for further information.

1.1 Harware Overview

EIP 7012 has the following interfaces for Networking, telephone interface, LED indication, and power connector. EIP 7012P supports POE that can connect to POE switch and use the power of POE switch.

Two RJ-45 Networking interface, these two interfaces support 10/100Mps Fast Ethernet. You can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.

LED Indication: some LED indicators in the VoIP Phone show the functions, like speaker phone, Register, and so on.

Dimension: approximately 18 x 20 x 8 cm

Weight: approximately 220g

Operating temperature: 0-50°C

Adapter: Input:100-240V, 50-60Hz

Output via general adapter: 6V, 1A

Output via Power over Ethernet on WAN port: Typical 5V, 1.2A (7012P only)

FCC Class B

CE, 89/336/EEC

LVD: 73/23/EEC

RoHS compliant

Warning: AC Input out of defined range will cause phone malfunction or damage!

1.2 Software Overview

Network Interface

- Two RJ45 10/100 Base-T Ethernet ports

Display

- 16 characters x 2 lines
- Backlit, grayscale

SIP

- SIP v2 (RFC 3261)
- SIP RPORT(RFC 3581)

- Many other SIP protocols
- Support 3 SIP accounts
- SIP server redundancy

Call Features

- Call Hold
- Call Waiting
- Call Forward
- Unconditional forward
- Busy forward
- Unavailable forward
- Call Transfer
- Blind transfer
- Attended transfer(Ring and Consult transfer)
- 3-way Conference
- Call Park/Call Pickup**
- On-demand Call Recording**
- Hotline

Phone Functions

- Feature keys
  - Phone Book
  - Redial
  - Do not disturb (DND)
  - Hold
  - Transfer
  - Conference
  - Forward
  - Voice Message
  - Speaker phone
  - Volume Adjustment
  - Call Recording
  - Mute
- Phone Book (140sets)
- Speed Dials (8 sets)
- Call History (64 sets)
- Caller ID Display
- Speaker Phone
- Multi-user(Up to 3)
- Mimic second dial tone
- Incoming call indicator
- MWI/VMWI
- Provide 4 different ringtones
  - Selectable ringtones
  - Caller ID ringtones
- Programmable tone
- Volume/Input gain adjustment
- Dial plan setting
  - Dial without waiting

DTMF

- In-band DTMF
- RFC2833(Out-of-band DTMF)
- SIP INFO

Codecs

- G.711 μ-law/A-law
- G.723.1 6.3k/5.3k bit/s
- G.726 16k/24k/32k/40k bit/s
- G.729A 8k bit/s
- G.729A/B
- GSM

Administration

- Web-based configuration
- Direct keypad configuration
- Auto Provisioning on boot**
- Automatic firmware upgrade**
- Firmware upgrade via web
- Diagnostics (Syslog support)

Voice Quality

- G.168 LEC (Line Echo Canceller)
- AEC (Acoustic Echo Canceller)
- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generator
- Packet loss compensation
- Adaptive jitter buffer

Networking

- IP assignments:
  - Static IP
  - DHCP
  - PPPoE
- STUN for NAT traversal
- MAC clone
- SNTP client
- Bridge/NAT mode on LAN
- DHCP server on NAT mode for LAN
- DDNS
- Virtual Server

Security

- HTTP 1.1 basic/digest authentication for web setup
- MD5 for SIP authentication (RFC 2069/RFC 2617)

QoS

- IEEE 802.1Q VLAN
- ToS field

2 Keypad Interface

2.1 Keypad Description

Key Name	Description
1	“1”, “-“, “,”, “!”, “?”
2	“2”, “a”, “b”, “c”, “A”, “B”, “C”
3	“3”, “d”, “e”, ”f”, “D”, “E”, “F”
4	“4”, “g”, “h”, “I”, “G”, “H”, “I”
5	“5”, “j”, “k”, “l”, “J”, “K”, “L”
6	“6”, “m”, “n”, “o”, “M”, “N”, “O”
7	“7”, “p”, “q”, “r”, “s”, “P”, “Q”, “R”, ‘S”
8	“8”, “t”, “u”, “v”, “T”, “U”, “V”
9	“9”, “w”, “x”, “y”, “z”, “W”, “X”, “Y”, “Z”
0	“0”, “space”
*	“*”, “•”, “:”, “@”
#	Start dialing process.
4-way Navigation Keys	Press to scroll through lists and menus on the display.
MENU	Press to access the menu options or cancel your selection and go back to the previous level.
ENTER	Press to enter a menu or confirm a selection.
Phone Book	Press to access the personal phonebook directory.
REDIAL	Press to call the last number dialed.
DND	Press to block all incoming calls.
HOLD	Press to put an active call on hold.
TRANSFER	Press to transfer an active call to another VoIP phone on the system.
CONFERENCE	Press to activate the three-way conference call.
FWD	Press to forward all incoming calls to another phone on the system.
DELETE	Press to erase the number you dialed when making a call.
M1~M8	Press any of the keys to speed dial the preset contact number.
M9	Press to enable the Mute function.

M10 / REC	Press to record the conversation, please refer to section 2.2 for further details.
VOICE MSG	Press to listen to voice mail messages.
SPEAKER	Press to activate the speakerphone to allow handsfree conversations.
VOLUMN Control Key	Press to increase or decrease the volume of the ringer tone, handset, or the volume of the current call using the speakerphone.

2.2 Recording

The recording function is for the user who wants to record the conversations during the calls. The user only needs to press “M10 / REC” button to start recording and press again to stop it. The voice file will be saved in the user’s voice mail system. The Message Waiting Indicator on IP phones will be lit to inform the user.

To listen to the recording, make a call to the user’s voice mail system, press “1” to hear the new messages of the recording.

2.3 Keypad Function and Setting List

2.3.1 Phone Book

Name	Description
Search	Search Phone Book.
Add Entry	Add new phone number to phone book.
Speed Dial	Add speed dial phone number to speed dial list.
Erase All	Erase all phone number from Phone Book.

The phone book feature enables you to create new entries for your contacts and edit their contact information. It also enables you to store a number from a received call or a dialed number in your phone book.

2.3.1.1 Creating a New Entry

To create a new entry, do the following:

1. Press the **Menu** key and then select **Phone book** -> **Add Entry** to add a new entry to the phone book.
2. Type the name of the new contact.
3. Press the **Enter** key to save the name.
4. Enter the phone number of the new contact.
5. Press the **Enter** key to save the number.

2.3.1.2 Modifying Contact Information

To modify contact information, do the following:

1. Press the **Menu** key and then select **Phone Book** -> **Search,** and press the **Enter** key to view the phone book entries.
2. Scroll through the phone to select an entry, and press the **Enter** key to display the contact information.
3. Press the **Enter** key to edit the name, and press it again to save your setting.
4. Change the phone number as required and press the **Enter** key to save your changes.

2.3.1.3 Deleting a Contact

You can delete all contacts in the phone book one at a time or all at once.

To delete a single contact, do the following:

1. Press the **Menu** key and then select **Phone book** -> **Search**.
2. Press the **Enter** key to display all contacts in the phone book.
3. Select the contact you want to delete and press the **Delete** key. The selected entry is deleted.

To delete all contacts in the phone book, do the following:

1. Press the **Menu** key and then select **Phone book**.
2. Choose **Erase All** and then use the up and down navigation keys to select **Yes** to confirm or **No** to cancel.

2.3.1.4 Speed Dial Setting

You can use speed dialing to assign a hotkey (M1 to M8) to the phone numbers that you dial most frequently. You can then press a hotkey (M1 to M8) to quickly dial a number instead of keying in the entire phone number or searching for it from the phone book.

To assign speed dial hotkeys, do the following:

1. Press the **Menu** key and then select **Phone book** and **Speed Dial**.
2. Input the phone number and press the **Enter** key. Then select one of the M1~ M8 keys to assign for speed dialing.

2.3.2 Call History

Name	Description
Incoming Calls	Show all incoming call.
Dialed Numbers	Show all dialed call.

Erase Records	Delete call history. **All:** Delete all call history. **Incoming:** Delete all incoming call. **Dialed:** Delete all dialed out call.

2.3.3 Phone Setting

Name	**Description**	
Call Forward	All Fwd.	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.
	Busy Fwd.	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.
	No Answer Fwd.	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.
	Ring Timeout	Set the Ring times to start the no answer forward function, ex: 2 means after 2 rings then forward to the dedicated number.
Do Not Disturb	Always	Block all phone calls.
	By Period	Block all phone calls at a certain period of time.
	Period Time	Set the start time and end time to Block Setting.
Alarm Setting	Activation	Set the Alarm Enabled or Disabled.
	Alarm Time	Set the time for alarming.
Date/Time	Date and Time Setting.	
	Date & Time	Set the IP Phone Date and Time.
	Time Format	To set the time as 12-hour or 24-hour clock.
	SNTP Setting	**SNTP:** Enabled / Disable SNTP. **Primary SNTP:** Set Primary SNTP server IP address. **Secondary SNTP:** Set Secondary SNTP server IP address. **Time Zone:** Set Time zone.
Volume & Gain	Handset Volume	Set Handset volume from 0~15 (max.) for you to hear.
	Speaker Volume	Set Speaker phone volume from 0~15 (max.) for you to

		hear.
	Handset Gain	Set Handset Gain from 0~15 (max.) for the other site to hear.
	Speaker Gain	Set Speaker phone Gain from 0~15 (max.) for the other site to hear.
Ringer	Ringer Volume	Ringer volume setting from 1~15 (max.).
	General Ringer	Ringer tone selection from 1~3 and default.
Digit Timeout	Set Auto Dial time from 1~5 seconds.	
Call Waiting	Set enable to be informed there is a new incoming call, user can set the function to disable.	
Direct Dial	Set enable to be able to dial the phone numbers in Phone Book by pressing “**+position”.	
Busy Redial	Set enable to redial the call up to 10 times when receiving SIP busy.	

2.3.4 Network

Name	**Description**	
WAN Setup	IP Type	**Fixed IP:** Self configure the IP address. **DHCP:** To get IP address through DHCP. **PPPoE:** To get IP address through PPPoE.
	Set Fixed IP	**IP Address:** The IP address of the IP phone. **Subnet Mask:** Configure subnet mask. **Default Gateway:** The gateway address
	Set PPPoE	**User Name:** User name of PPPoE **Password:** Password of PPPoE
LAN Setup	Bridge	Set LAN as bridging mode
	NAT	Set LAN as NAT mode
DNS	Primary DNS	First DNS address
	Secondary DNS	Second DNS address
VLAN	Activation	Activate or disable the VLAN.
	VID	Set VID from 2 to 4094.
	Priority	Set the priority from 0 to 7.

	CFI	0~1
Status	Show WAN, LAN IP address and MAC address.	

2.3.5 SIP Setting

If you want to use keypad to set the SIP setting, you have to go to item 8 Sys. Authority to input the password, or you can not change the SIP setting.

Name	**Description**	
Service Domain	First/Second/Third Realm	The realms include following information. You can press “1*#”, “2*#”, and ”3*#” to change among these three SIP realms. **Activation:** To enable or disable the realm. **User Name:** The SIP’s user name. **Display Name:** The SIP’s display name. **Register Name:** The SIP’s registered name. **Register Pwd.:** The SIP’s password. **Proxy Server:** The address of SIP proxy. **Domain Server:** The address of domain server. **Outbound Proxy:** The address of outbound proxy.
Codec	Codec Type	The codec type includes G.711 uLaw, G.711 aLaw, G.723, G.729, G.726-16, G.726-24, G.726-32, and G.726-40.
	VAD	Voice Active Detection Enable/Disable.
RTP Setting	Outband DTMF	Enable/Disable outband DTMF.
	Duplicate RTP	**No Duplicate:** Do not resend the voice packages. **One Duplicate:** Resend voice packages one time. **Two Duplicate:** Resend voice packages two times.
RPort Setting	RPort Enabled/Disabled	
Hold by RFC	Enable/Disable Holding the calls, according to RFC3261.	
Status	Show the SIP Proxy register status. You can use UP/Down key to check each Realm’s status.	

2.3.6 NAT Traversal

If you want to use keypad to set the NAT Traversal settings, you have to go to item 8 Sys. Authority to input the password, or you can not change the NAT Traversal settings.

Name	Description	
STUN Setting	STUN	Enable/Disable STUN.
	STUN Server	The address of STUN server.

2.3.7 Administrator

If you want to use keypad to set the Administrator setting, you have to go to item 8 Sys. Authority to input the password, or you can not change the Administrator setting.

Name	Description	
Upgrade System	This function must work with IP PBX.	
	Upgrade Now	Select to direct connect to IP PBX to check if there is any upgrade version. If there is a newer version, IP phone will upgrade the system automatically
	Schedule State	Select to see the current status and scheduling time.
Set To Default	You can restore to the default setting.	
Change Pwd.	Press ENTER and press a new password to replace the original password with the new one.	
Version	This will show the system's firmware version.	
Vendor ID	To see the vendor ID of the IP Phone	
Watch Dog	You can use this to enable Watch Dog function to do the debugging.	
Restart	You can use this function to restart your IP Phone.	

2.3.8 Sys. Authority

To do the SIP setting, NAT traversal and Administrator from Keypad, you need to input the password first. Default is "test".

3 Setup the VoIP Phone by Web Browser

Default the IP Phone's Bridge is enabled, WAN port is in DHCP Client Mode. LAN port is the same as WAN. Connect the VoIP Phone to DHCP server, and the server assign an IP address to the phone. Check the IP address and add the port number ":9999" at the end of the IP address to access the web browser.

Note: When selecting DHCP Client and cannot detecting the IP, the VoIP Phone will provide a default IP, 192.168.2.25.

Note: It is highly recommended to use Internet Explorer 6.0 or later for web configurations.

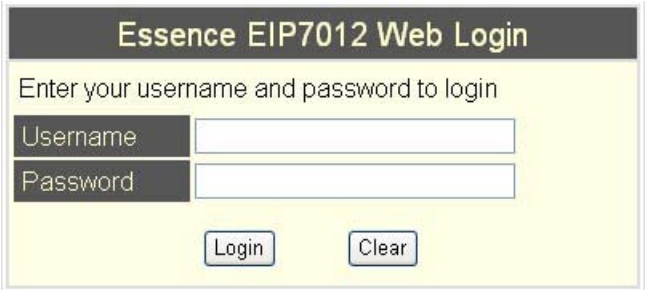
3.1 Login

Please input the username and password into the blank field. The default setting is:

1. For Administrator, the username is: *root*; and the password is: *test*. If you use the account login, you can configure all the setting.
2. For normal user, the username is: *user*; and the password is: *test*. If you use the account login, but you can not configure the SIP setting.

Click **Login** to move into the VOIP PHONE web based management information page.

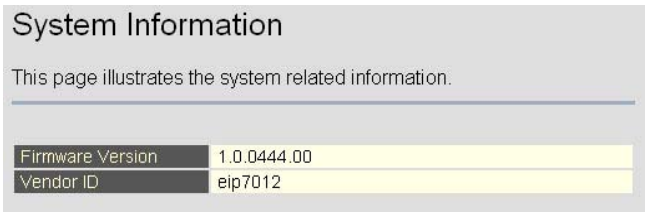
Any changes in the Web Management interface except for Phone Book require clicking **Submit** in that page, and then go to the Save Changes page and click **Save**. The system will restart, and all the settings can work properly.



3.2 System Information for the VoIP Phone

When you login the web page, you can see the current system information of the VoIP Phone like Model Name, Firmware Version, Vendor ID and Codec Version in this page.

Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.



Note: The information in this page is useful when you need technical support. You may screen capture this page and provide the information to technical support team so that they could easily track the issues you have.

3.3 Phone Book

Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial number. The Phone Book can store 140 phone numbers and the Speed Dial can store 8 phone numbers. If you want to use Speed Dial you just dial the speed dial number then press "#".

3.3.1 Phone Book

In the Phone Book function, you can add/delete the phone number in the phone book list. You can add maximum 140 entries phone book list.

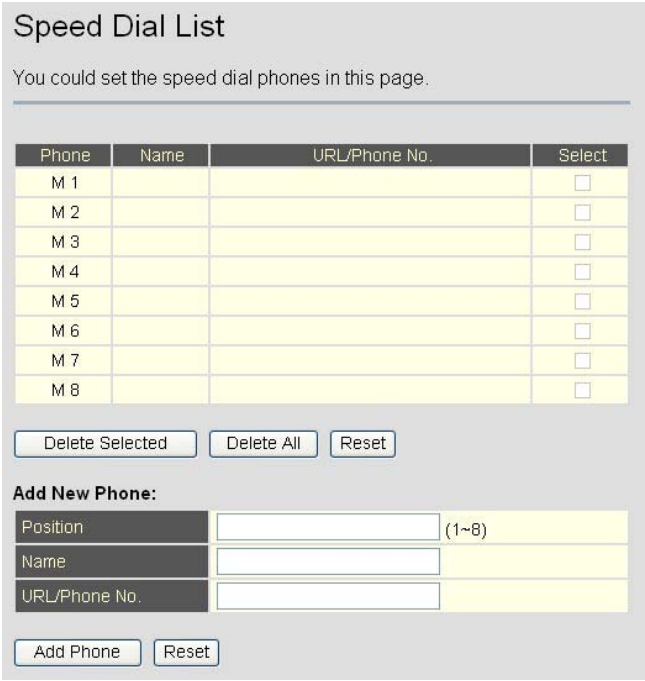
If you need to add a phone number into the phone book, you need to type the position, the name, and the phone number[1] and ringer type for the number. When you finished a new phone list, just click **Add Phone**.

If you want to delete a phone number, you can select the phone number you want to delete then click **Delete Selected**.

If you want to delete all phone numbers, you can click **Delete All**.

3.3.2 Speed Dial List

In Speed Dial List function, you can add/delete Speed Dial number. You can input maximum 8 entries speed dial list.



If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click **Add Phone**.

If you want to delete a phone number, you can select the phone number you want to delete then click **Delete Selected**.

If you want to delete all phone numbers, you can click **Delete All**.

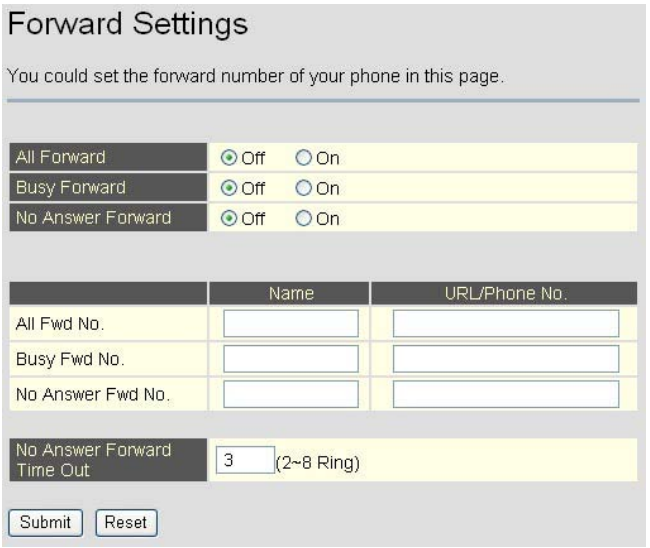
[1] The format of the phone number here can either be phone numbers or SIP URL format. General phone numbers will work in most cases if VOIP phone is deployed under IP PBX environment. For direct IP dialing, a SIP URL format can be also used.

3.4 Phone Setting

Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Ringer Settings, DND Settings, Dial Plan Settings, Call Waiting Settings, Soft-key Setting functions, Hot Line Settings, Alarm Settings and Daylight Settings.

3.4.1 Forward Settings

You can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by clicking the icon.



3.4.1.1 All Forward

All incoming calls will forward to the number you entered. You can input the name and the phone number in the field accordingly.

3.4.1.2 Busy Forward

If you are on the phone, the new incoming call will forward to the number you entered. You can input the name and the phone number in the **URL/Phone No.** box.

3.4.1.3 No Answer Forward

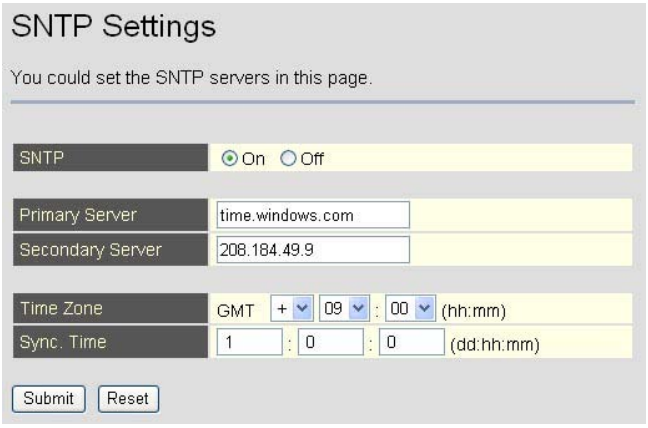
If you can not answer the phone, the incoming call will forward to the number you entered. You can input the name and the phone number in the **URL/Phone No.** box. Also you have to set the Time Out time for system to start to forward the call to the number you select.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to

reflect the changes.

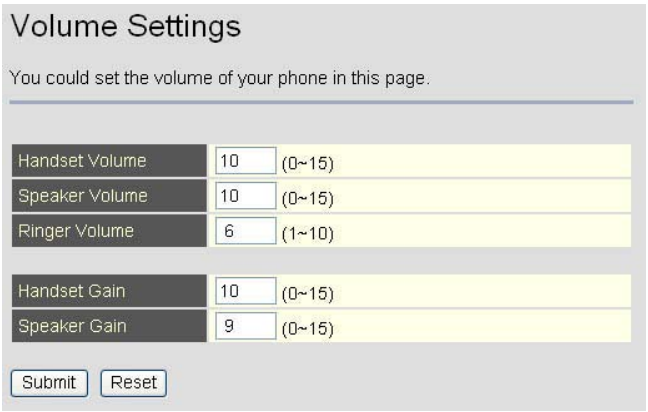
3.4.2 SNTP Settings

You can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.4.3 Volume Settings

You can setup the Handset Volume, Speaker Volume, Ringer Volume, the Handset Gain, and Speaker Gain.



Handset Volume is to set the volume you hear from the handset.

Speaker Volume is to set the volume you hear from the speaker phone.

Ringer Volume is to set the ringer volume.

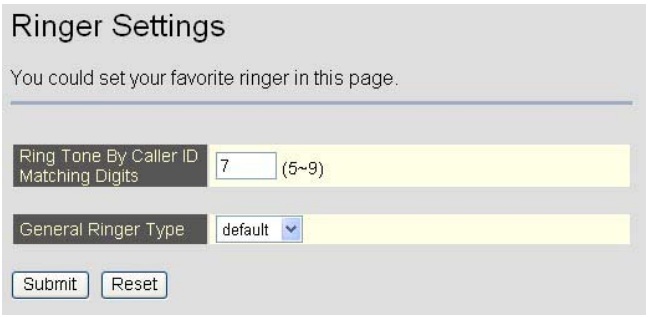
Handset Gain is to set the volume send out from the handset.

Speaker Gain is to set the volume send out from the micro phone.

When finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.4 Ringer Settings

You can select the melody for the incoming calls. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



Ring Tone By Caller ID Matching Digits: Enter a digit for the number of caller ID digits. If the total digit of the phone number is larger or equal to the digit, the ringer type hunts for the phone number that matches digits at the end, and the melody of that phone number will ring. If the total digit of the phone number is smaller than the digit, it will hunt for the exact match phone number and ring the ringer type of the phone number.

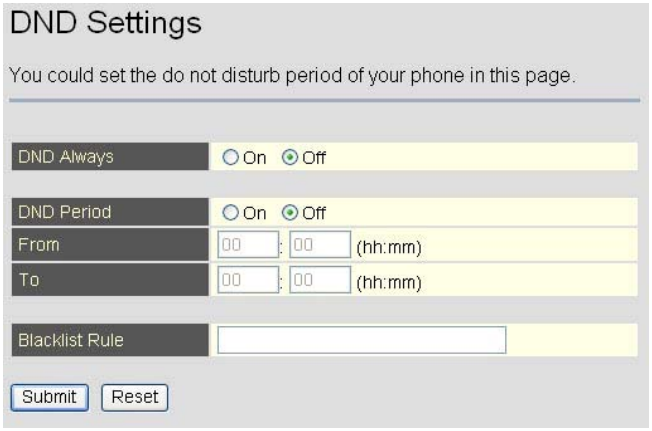
Note: The order of the phone number in the Phone Book page matters the hunting sequential.

General Ringer Type: Select general ring tone for the incoming calls.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.5 DND Settings

You can setup the DND (Do-not-disturb) Setting to keep the phone silence. You can choose **DND Always** or **DND Period**.



DND Always: All incoming call will be blocked until disable this feature.

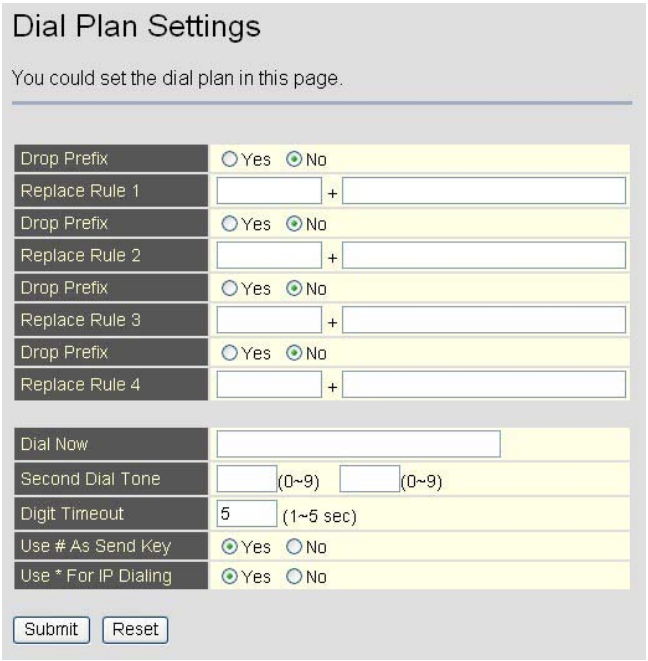
DND Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.

Blacklist Rule: Enter a rule[2] to block calls that fit the rule.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.6 Dial Plan Setting

This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.



[2] For how to configure the rule, please refer to Dial Plan Setting.

3.4.6.1 Symbol Explain

x or X	0,1,2,3,4,5,6,7,8,9
+	or

Drop Prefix: The default is **No** for adding the prefix to the dial plan in **Replace Rule**. Click **Yes** to delete the prefix of the dial plan in **Replace Rule**.

Replace Rule 1~4: Type rule for matching.

Example:

Drop Prefix	Replace Rule		Description
No	002	8613+8662	When the dialed number begins with “8613” or “8662”, the number will be added “002” to call out.
No	002	12	When the dialed number begins with “12”, the number will be added “002” to call out.
No	002	5xxx	When the dialed number begins with “5”, and has total of four digits, the number will be add “002” to call out.
Yes	002	003+004+005	When the dialed number begins with “003”, “004” or “005”, the number will be dropped “003”, “004” or “005”, and then add “002” to call out.
Yes	002	55xxxx	When the dialed number begins with “55” and has total of six digits, the number will be dropped “55”, and add “002” to call out.

Dial Now: Type a number or dial pattern. When the dialed number match the patter, the phone will directly send out the number. Note that the first digit cannot set “0”.

Second Dial Tone: Press the entered number to hear the second dial tone.

Digit Timeout: The default is 5 seconds. Type a time in sec for auto dial after the time is up.

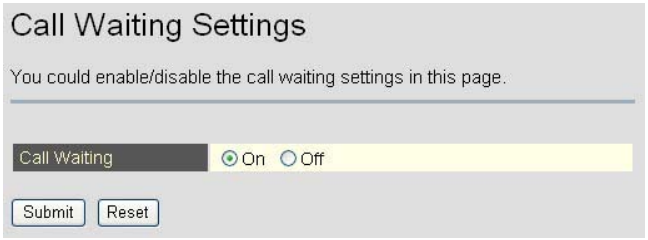
Use # As Send Key: The default is **Yes**. Click **No** to disable send key function, and the dialed number will not send out until Auto Dial Time is finished.

Use * For IP Dialing: The default is **Yes**. Click **No** to see “*” (star) as it is instead of “.” (dot).

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.7 Call Waiting Settings

If the user doesn't want to be informed there is a new incoming call, the user can set the function off. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.4.8 Soft-key Settings

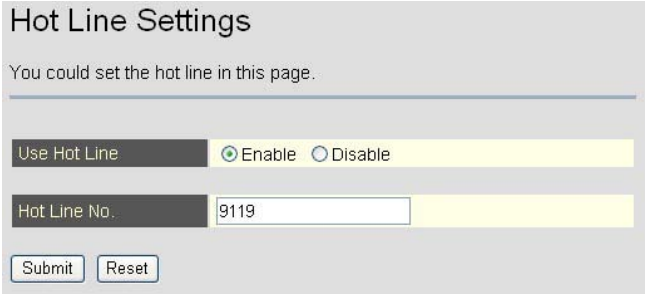
EIP 7012 supports soft-key settings for voice messages. You can press VOICE MSG on the VoIP Phone to dial out the entered number in order to get into the voice mail service.

Voice mail key: Type a serial of number for listening to the voice messages.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.9 Hot Line Settings

Hot line setting is a special line for the VoIP Phone. When enable the function, the phone will directly dial out the number once the phone is picked up.

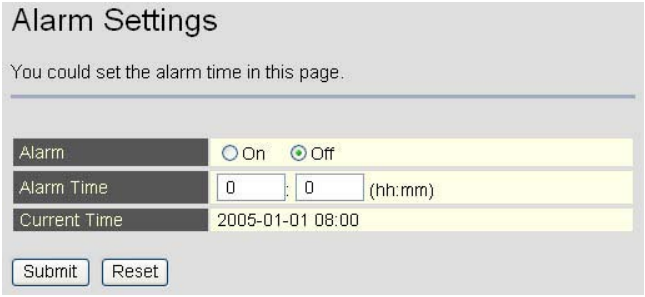


Hot Line Number: Type a phone number as a special line.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.10 Alarm Settings

Alarm Settings is to inform user at a certain time. Click **On** and then type the time for ringing.



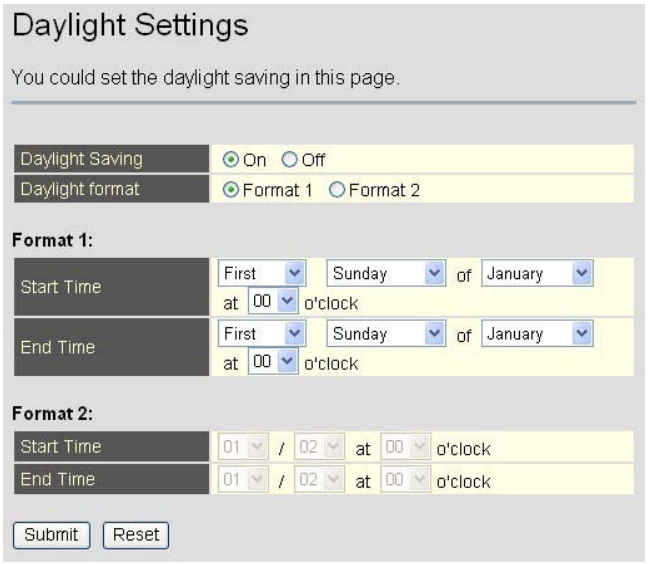
Alarm Time: Type a time for reminding.

Current Time: Display the current time of the VoIP Phone.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.4.11 Daylight Settings

If the country has daylight saving time, you can configure the time in this page.



Click **On**, select **Format 1** or **Format 2**, and configure the start/end time to enable daylight saving at the configured period of time.

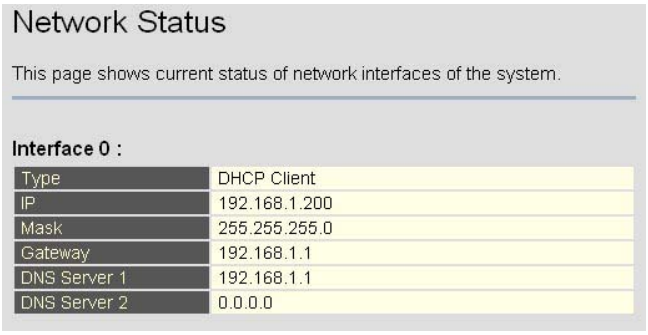
When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.5 Network

In Network, you can check the Network status, configure the WAN Settings, LAN Settings, DDNS settings, VLAN Settings, DMZ Settings, Virtual Server and PPTP Settings.

3.5.1 Network Status

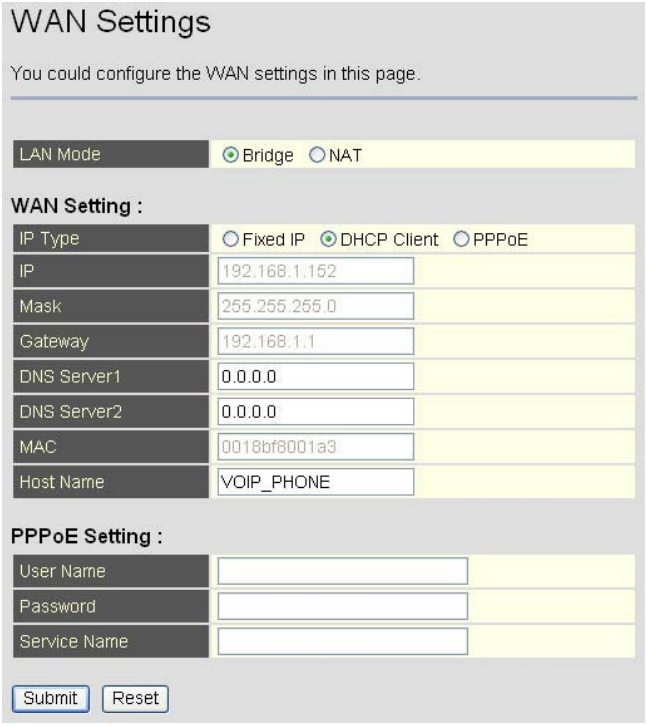
You can check the current Network settings in this page.



3.5.2 WAN Settings

In this page, you can configure the VoIP Phone's WAN port setting. The WAN port is for you to connect

to the ADSL Router, or Broadband Router. Also, you can use PPPoE to get the WAN IP address from your ISP.



The default setting is **Bridge** mode. If you don't need to use the Bridge mode, you can change to **NAT** mode.

The WAN port default is **DHCP Client** mode. You can change the setting to **Fixed IP** mode, or **PPPoE** mode.

If you change the WAN port's setting to **Fix IP** mode, then you have to make sure the IP address. Net Mask, Gateway, and DNS settings are suitable in your current network environment.

Host Name: Name of the phone. The information can be transmitted in a packet.

If you change the WAN port's setting to **PPPoE** mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.

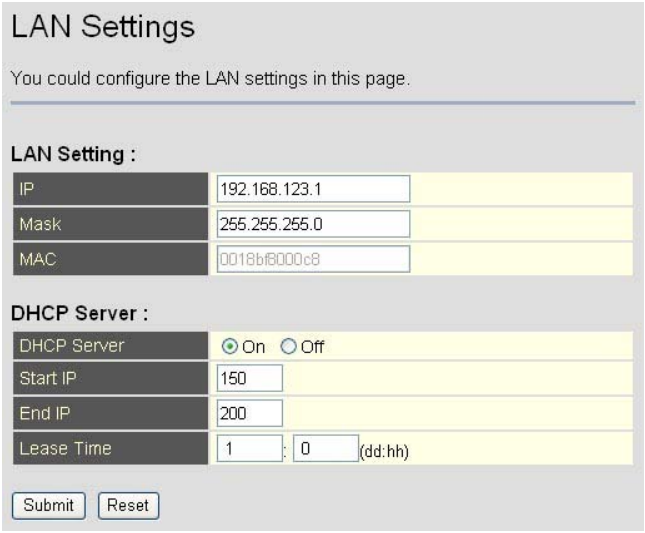
Service Name: Name of the server. You can contact your SIP provide for the information.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

Note: When selecting DHCP Client and cannot detecting the IP, the IP Phone will provide a default IP, 192.168.2.25.

3.5.3 LAN Settings

In this page, you can configure the VoIP Phone LAN port's setting.



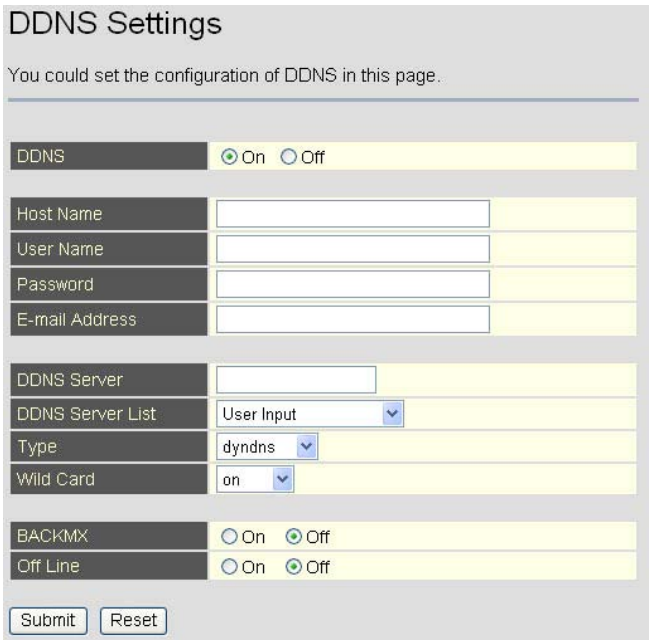
The LAN settings' default does not effect as **Bridge** mode is enabled. To configure LAN settings, go to **Network** -> **WAN Settings**, and click **NAT** under **LAN Mode**.

You can connect your PC to the LAN port, set your PC as DHCP Client mode, and then you can get IP address from the VoIP Phone.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.5.4 DDNS Settings

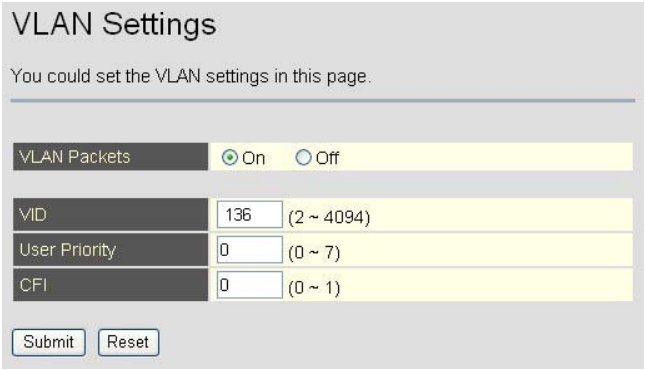
You can configure the DDNS setting in this page. You need to have the DDNS account and input the information properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.5.5 VLAN Settings

You can set the VLAN setting in this page.

There are two kind of destination packets coming from the VoIP Phone's WAN port. One packet goes to the VoIP Phone, and the other goes through the LAN port to the PC.



VLAN Packets: If you enable the first VLAN Packets and set the VID, User Priority, and CFI, all the incoming packets will be check with the IP Address and the VID.

VID: You can follow your service provider to set your VID.

User Priority: Defines user priority, giving 8 priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.

CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.

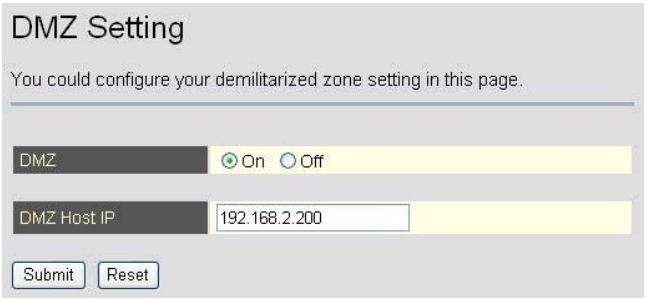
When you enable the first VLAN Packets and set the VID, User Priority, and CFI, all the incoming packets with the VoIP Phone's IP address and the same VID will be accept by the VoIP Phone. If the incoming packets with the VoIP Phone's IP address but the different VID then the packets will be discard by the VoIP Phone. The Other incoming packets with different IP address will go through the LAN port to the PC.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.5.6 DMZ Setting

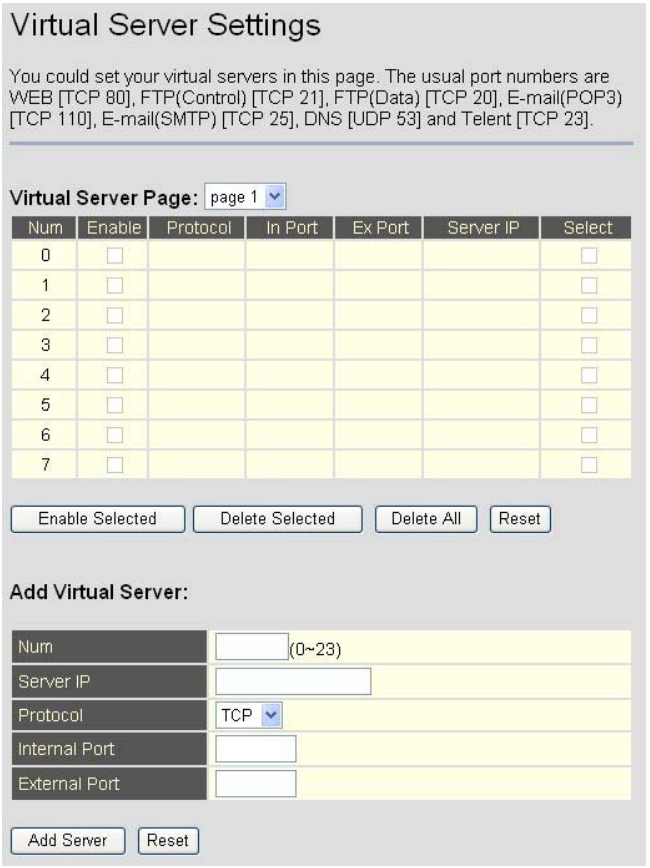
You can enable DMZ Setting of the VoIP Phone. Click **On**, and type an IP address of the PC in **DMZ Host IP**.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.5.7 Virtual Server

EIP 7012 provides configuring as a virtual server function.


Virtual Server Settings
You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP(Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telent [TCP 23].
Virtual Server Page: page 1
Num Enable Protocol In Port Ex Port Server IP Select
0 1 2 3 4 5 6 7
Enable Selected
Delete Selected
Delete All
Reset
Add Virtual Server:
Num (0~23)
Server IP
Protocol TCP
Internal Port
External Port
Add Server
Reset

If you need to add a virtual server into the Virtual Server list, you need to type the num, server IP, internal/external port, and select a protocol. When you finished a new phone list, just click **Add Server**.

If you want to enable a virtual server, you can select the virtual server you want to enable, and then click **Enable Selected**.

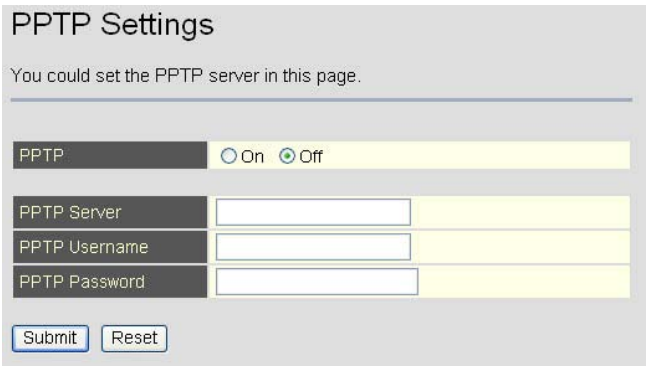
If you want to delete a virtual server, you can select the virtual server you want to delete, and then click **Delete Selected**.

If you want to delete all virtual servers, you can click **Delete All**.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.5.8 PPTP Settings

PPTP is one kind of VPN. Remote users can use local telephone number and network through ISP. Click **On** and type the server, user name and password of PPTP to enable the function. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



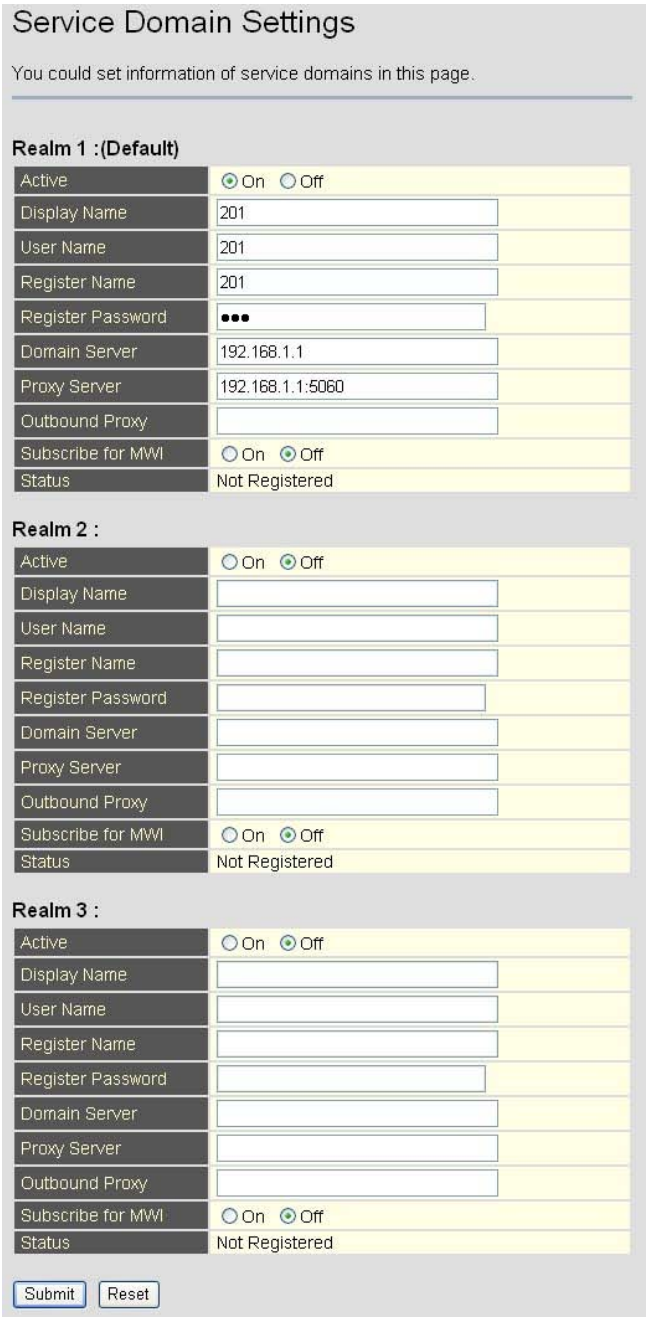
Note: You need to configure **LAN Mode** as **NAT** in WAN Settings page to enable this function.

3.6 SIP Settings

In SIP Settings, you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related information correctly then you can register to the SIP Proxy Server correctly.

3.6.1 Service Domain

In Service Domain Function, you need to input the account and the related information in this page. Please refer to your ISP provider. You can register three SIP account in the VoIP Phone. You can dial the VoIP phone to your friends and receive the phone from any of these three SIP accounts.



First you need click **On** in **Active** to enable the Service Domain, and you can input the following items:

Display Name: You can input the name you want to display.

User Name: You need to input the User Name get from your ISP.

Register Name: You need to input the Register Name get from your ISP.

Note: Usually User Name and Register Name are the same unless specified by ISP.

Register Password: You need to input the Register Password get from your ISP.

Domain Server: You need to input the Domain Server get from your ISP.

Note: The default SIP port of the Domain Server is 5060. If you want to change the SIP port, specify the port here.

Example: 192.168.1.100 (Assume Domain server SIP port = 5060)

192.168.1.100:5678 (change the SIP port to 5678)

Proxy Server: You need to input the Proxy Server get from your ISP.

Outbound Proxy: You need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

Subscribe for MWI: You can click **On** for the VoIP Phone to ask for MWI periodically.

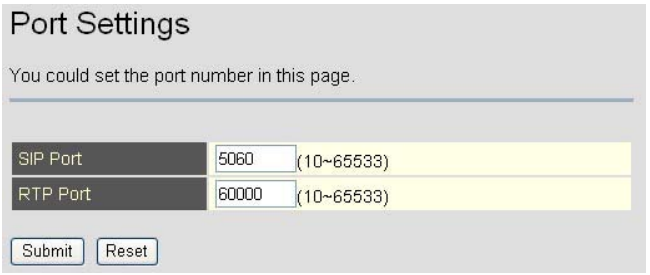
You can see the Register Status in the Status item. If the item shows “Registered”, then your VoIP Phone is registered to the ISP, you can make a phone call directly.

If you have more than one SIP account, you can follow the steps to register to the other ISP.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

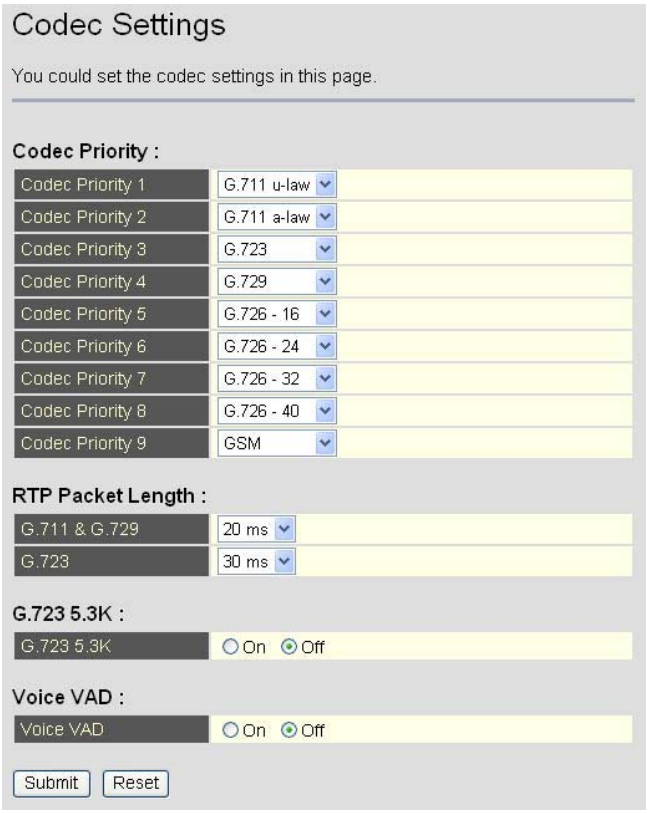
3.6.2 Port Settings

You can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTP port setting, please refer to the ISP to setup the port number correctly. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



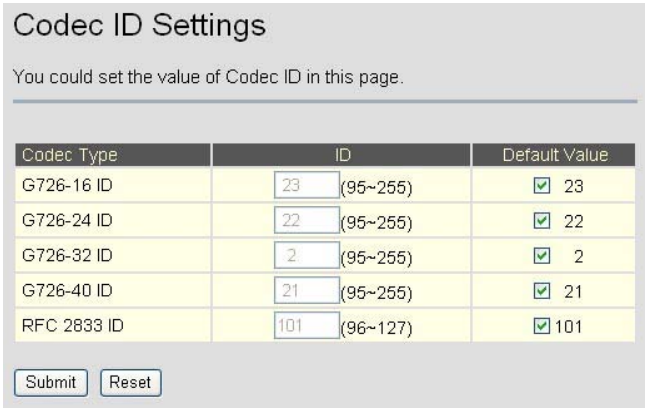
3.6.3 Codec Settings

You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.


Codec Settings
You could set the codec settings in this page.
Codec Priority :
Codec Priority 1 G.711 u-law
Codec Priority 2 G.711 a-law
Codec Priority 3 G.723
Codec Priority 4 G.729
Codec Priority 5 G.726 - 16
Codec Priority 6 G.726 - 24
Codec Priority 7 G.726 - 32
Codec Priority 8 G.726 - 40
Codec Priority 9 GSM
RTP Packet Length :
G.711 & G.729 20 ms
G.723 30 ms
G.723 5.3K :
G.723 5.3K On Off
Voice VAD :
Voice VAD On Off
Submit Reset

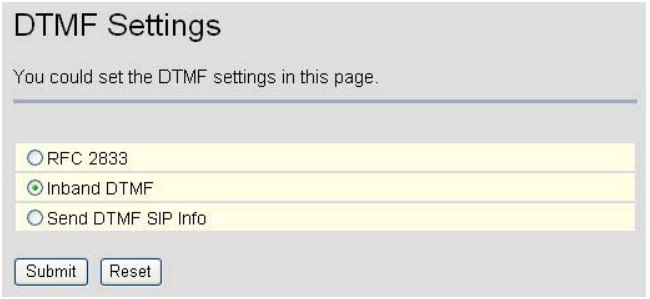
3.6.4 Codec ID Settings

You can set the Codec ID to meet the other device's requirement. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.


Codec ID Settings
You could set the value of Codec ID in this page.
Codec Type ID Default Value
G726-16 ID 23 (95~255) 23
G726-24 ID 22 (95~255) 22
G726-32 ID 2 (95~255) 2
G726-40 ID 21 (95~255) 21
RFC 2833 ID 101 (96~127) 101
Submit Reset

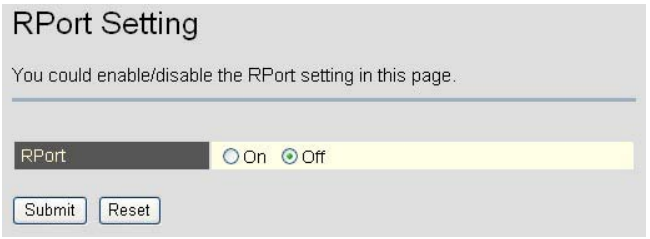
3.6.5 DTMF Settings

You can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please follow your ISP information. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.6.6 RPort Function

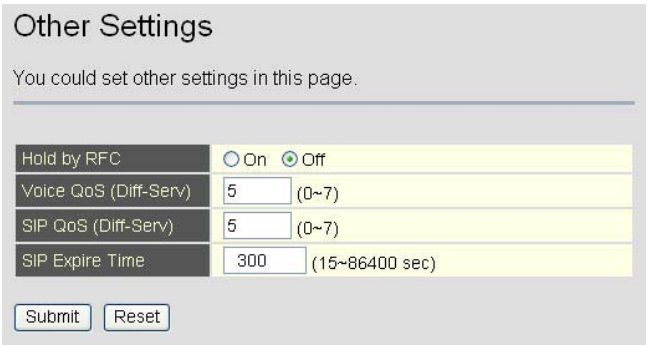
You can setup the RPort Enable/Disable in this page. To change this setting, please follow your ISP information. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.6.7 Other Settings

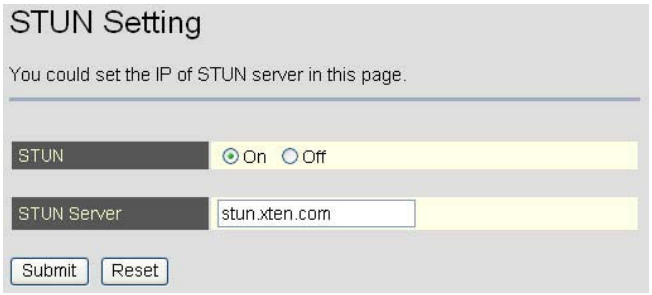
You can setup the Hold by RFC, Voice/SIP QoS and SIP Expire Time in this page. To change these settings please follow your ISP information. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. Note that the QoS function still needs to cooperate with the others Internet devices.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.7 NAT Trans

In NAT Trans, you can setup STUN[3] function. These functions can help your VoIP Phone working properly behind NAT.



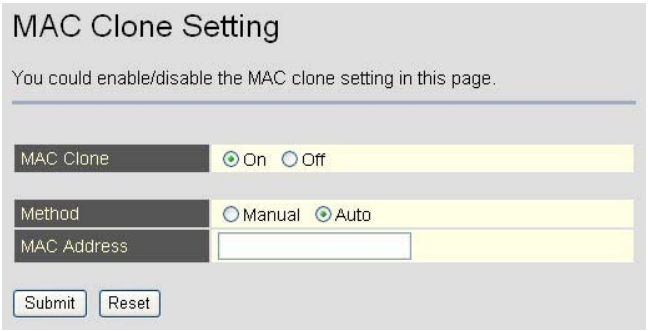
STUN Setting: You can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP Phone working properly behind NAT. To change these settings please following your ISP information. When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.8 Others

In Others, you can setup MAC Clone Setting, Tones Settings and Advanced Settings.

3.8.1 MAC Clone Setting

When connecting to ISP vial PPPoE, MAC Clone function can copy the PC's MAC address to IP.



Method: You can manually type a MAC address or let the VoIP Phone automatically detect the PC's MAC.

[3] **STUN (Simple Traversal of UDP (User Datagram Protocol) through NATs (Network Address Translators)** is a network protocol allowing a client behind a NAT (or multiple NATs) to find out its public address, the type of NAT it is behind and the internet side port associated by the NAT with a particular local port. This information is used to set up UDP communication between two hosts that are both behind NAT routers. The protocol is defined in RFC 3489.

MAC Address: When click **Manual**, type an MAC address for WAN port.

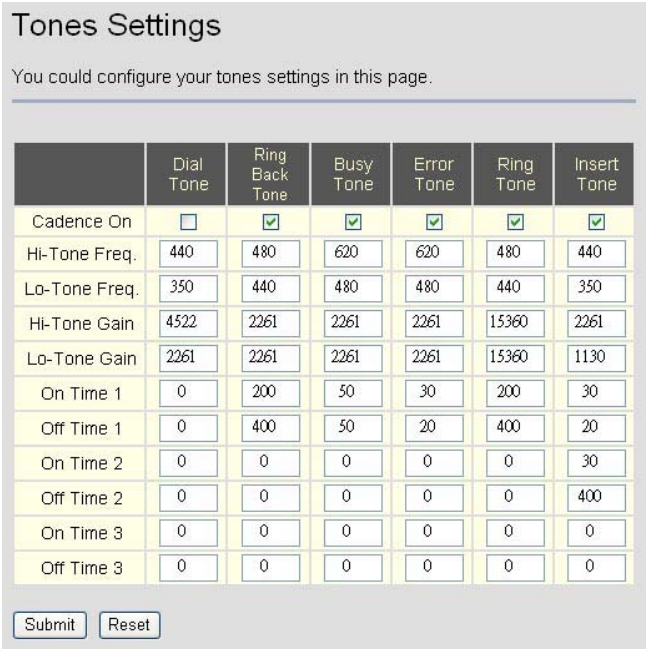
When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.8.2 Tones Settings

Tones Settings function can let you to configure the tones of various sounds.

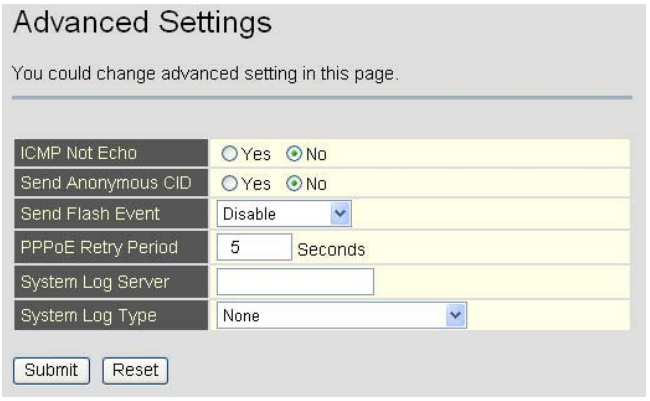
Unlike traditional phones, VoIP phones generate the tones for different state by itself. The standard of tones is different from country to country, you can customize the tone settings here.

When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.



3.8.3 Advanced Settings

You can configure some more advanced settings such as **ICMP Not Echo**, **Send Anonymous CID**, **Send Flash Event**, **PPPoE Retry Period**, **System Log Server** and **SIP Log Type** in this page.



ICMP Not Echo: Setup the ICMP echo Enable/Disable. This function can disable echo when someone pings this device, it can avoid hacker try to attack the device.

Send Anonymous CID: click **Yes** for not sending the phones caller ID out.

Send Flash Event: click **DTMF EVENT** or **SIP INFO** for the flash event.

PPPoE Retry Period: Type the time in second for PPPoE to redial, if connection is failed.

System Log Server: Type a server's IP address for saving system log.

System Log Type: Select types of log for saving. The phone offers different debugging levels, each levels contains different sets of debugging messages to send.

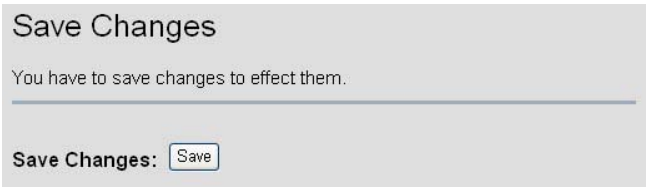
When you have finished the settings, click **Submit**. Go to the Save Changes page and click **Save** to reflect the changes.

3.9 System Auth

In System Authority, you can change your login name and password.

3.10 Save Changes

In Save Changes, you can save the changes you have done. If you want to use new setting in the VoIP Phone, You have to click **Save**. After you click **Save**, the VoIP Phone will automatically restart and the new settings will take effect.

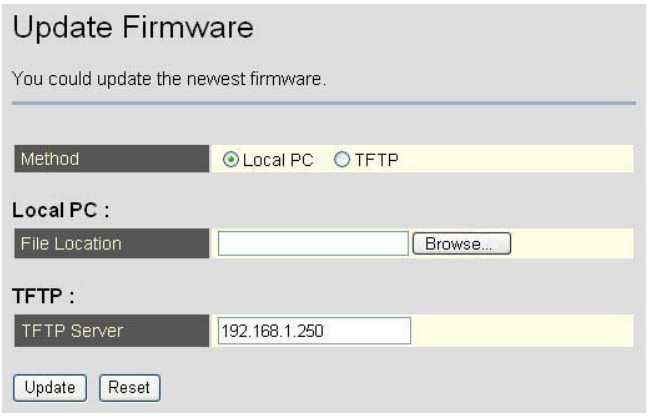


3.11 Update

In Update, you can update the VoIP Phone's firmware to the new one or do the factory reset to let the VoIP Phone back to default setting.

3.11.1 Update Firmware

In New Firmware function you can update new firmware via HTTP in this page. You can upgrade the firmware form your local PC or TFTP.



Update from Local PC:

Click **Browse** at the right side of the File Location to select a firmware or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the VoIP Phone, and then click **Update**.

Update from TFTP:

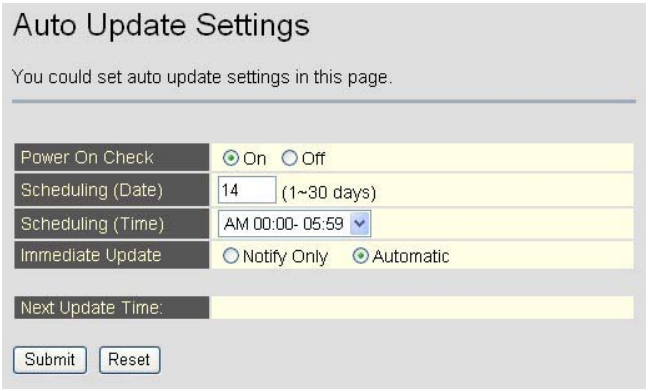
Type the address of the TFTP server, and then click **Update**.

Note: Do not change the firmware file name, otherwise the system will reject it.

Note: For TFTP server must contain updatelist.dat which reveals the intended update filename.

3.11.2 Auto Update Settings

The VoIP Phone provides automatic update function. Once enable the function, the phone will check the updates at the time in order to have the latest version of firmware. Note that the function must be in DHCP Client mode.



Note: Please check if the IP PBX supports the function.

Power On Check: click **On** to check if there is any new firmware when the power of VoIP phone is on.

Scheduling (Date): Type a duration in days for the phone to check the firmware.

Scheduling (Time): Select a period of time for updates.

Immediate Update: When the VoIP Phone detects there is newer version of firmware, you can select either to send out a notification only or directly update to the newer version.

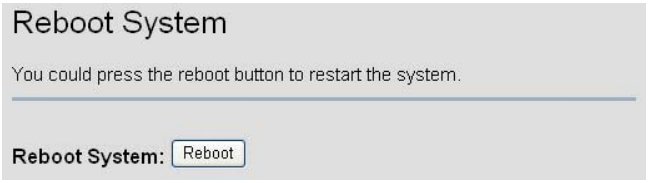
3.11.3 Restore Default Settings

In Default Settings, you can restore the VoIP Phone to factory default in this page. You can just click **Restore**, and the VoIP Phone will restore to default and automatically restart again.

Restore Default Settings
You could click the restore button to restore the factory settings.
Restore Default Settings:
Restore

3.12 Reboot

Reboot function is to restart the VoIP Phone. If you want to restart the VoIP Phone, you can just click **Reboot**, then the VoIP Phone will automatically.


Reboot System
You could press the reboot button to restart the system.
Reboot System:
Reboot

4 Automatic Client Configuration with Essence IP PBX

Auto Client Configuration (ACC) function can be used to download the original configurations stored in the Essence IP-PBX. This is very useful for the administrator who needs to set large amount of VoIP phones. The administrator can set a new user account in the IP PBX web page. Once the VoIP phone is connected to the IP PBX, it automatically downloads a predefined configuration setting from Essence IP-PBX.

ACC function is enabled in IP-Phone default settings. User does not need to change any settings in EIP 7012.